User Manual

HandyTone-496

Analog Telephone Adaptor

For SW Release Version 1.0.3.44

Grandstream Networks, Inc.

www.grandstream.com

GRANDSTREAM NETWORKS
Grandstream Innovative IP Telephony

Table of Contents

1   Welcome

Congratulations on becoming an owner of HandyTone-496. You made an excellent choice and we hope you enjoy all of its capabilities.

Grandstream's HandyTone-496 is an all-in-one VoIP integrated access device that features superb audio quality, rich functionalities, high level of integration, compactness and ultra-affordability. The HandyTone-496 is fully compatible with SIP industry standard and can interoperate with many other SIP compliant devices and software on the market.

Grandstream HandyTone-496 is a new addition to the popular HandyTone product family. The new HandyTone-496 features two FXS ports each with independent SIP accounts.

This document is subject to changes without notice. The latest electronic version of this user manual is available for download from the following location:

> http://www.grandstream.com/user_manuals/ HT496UserManual.pdf
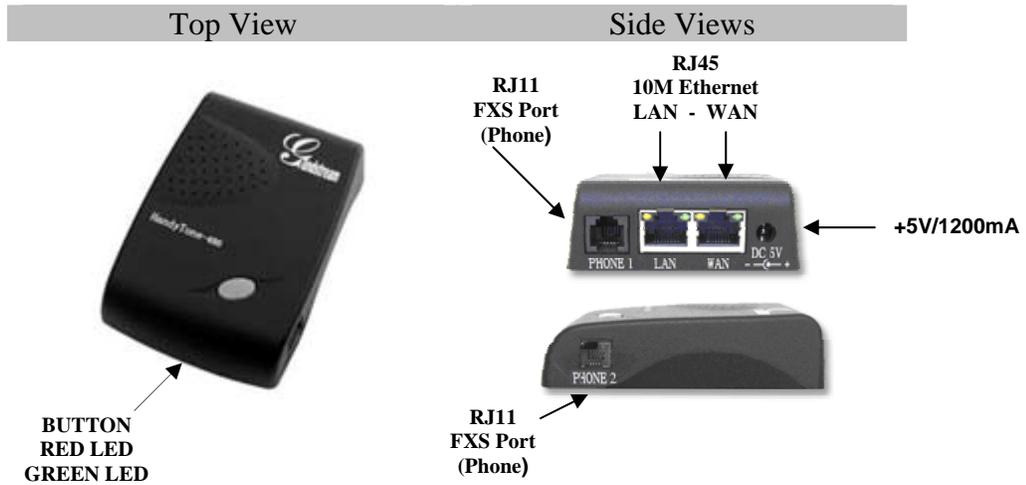
2   Installation

HandyTone-496 Analog Telephone Adaptor is an all-in-one VoIP integrated device designed to be a total solution for networks providing VoIP services.

The HandyTone-496 VoIP functionalities are available via regular analog telephones.

The following photo illustrates the appearance of a HandyTone-496.

Interconnection Diagram of the HandyTone-496:
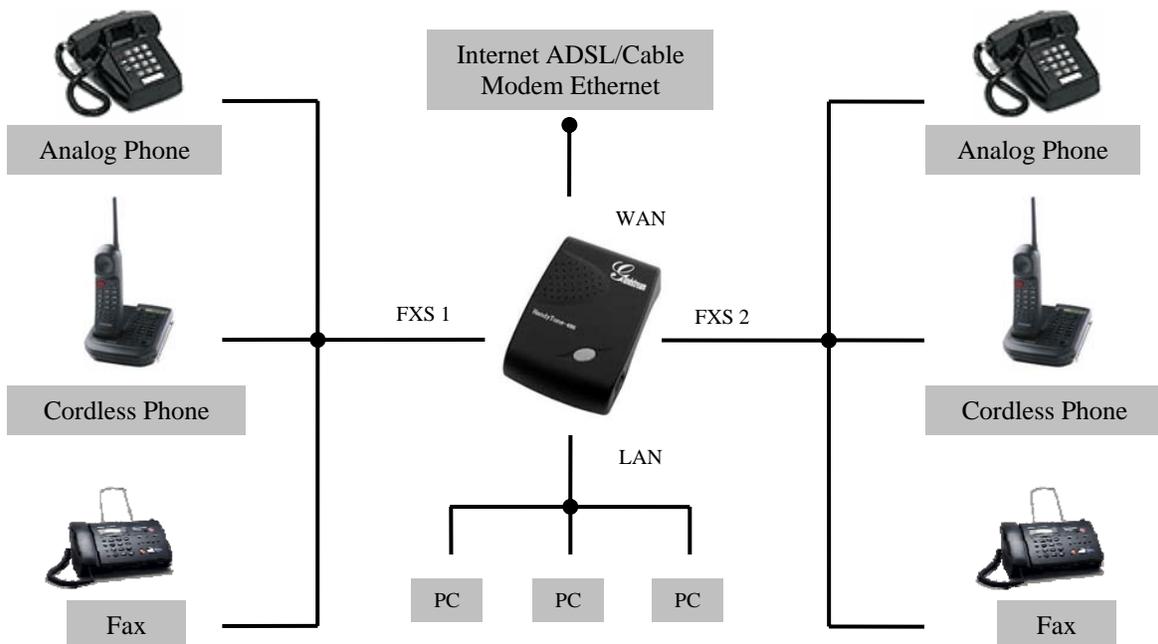
HandyTone-496 has two FXS ports. The PHONE1 port next to the LAN is FXS port 1 and the PHONE2 on the side of the HandyTone-496 is FXS port 2. Each FXS port can have a separate SIP account. This is a key feature of HandyTone-496. Both ports can make calls concurrently.

Following are the steps to install a HandyTone-496:

1. Connect a standard touch-tone analog telephone (or fax machine) to PHONE1 port.
2. Connect another standard touch-tone analog telephone (or fax machine) to PHONE2 port.
3. Insert the Ethernet cable into the WAN port of HandyTone-496 and connect the other end of the Ethernet cable to an uplink port (a router or a modem, etc.)
4. Connect a PC to the LAN port of HandyTone-496.
5. Insert the power adapter into the HandyTone-496 and connect it to a wall outlet.

Please follow the instructions in section 6.2.1 to configure the HandyTone-496.

3   What is Included in the Package

The HandyTone-496 package contains:

    1) One HandyTone-496
    2) One universal power adaptor
    3) One Ethernet cable

3.1   Safety Compliances

The HandyTone-496 is compliant with various safety standards including FCC/CE and C-tick. Its power adaptor is compliant with UL standard. The HandyTone-496 should only operate with the universal power adaptor provided in the package.

3.2   Warranty

Grandstream has a reseller agreement with our reseller customer. End users should contact the company from whom you purchased the product for replacement, repair or refund.

If you purchased the product directly from Grandstream, contact your Grandstream Sales and Service Representative for a RMA (Return Materials Authorization) number.

Grandstream reserves the right to remedy warranty policy without prior notification.

Warning: Please do not attempt to use a different power adaptor. Using other power adaptor may damage the HandyTone-496 and will void the manufacturer warranty.

Caution: Changes or modifications to this product not expressly approved by Grandstream, or operation of this product in any way other than as detailed by this User Manual, could void your manufacturer warranty.

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4   Product Overview

4.1   Key Features

- Supports SIP 2.0(RFC 3261), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), NTP, PPPoE, STUN, TFTP, etc.
- Built-in router, NAT, Gateway and DMZ port forwarding
- Device bridge mode support
- Supports dual SIP accounts via dual FXS ports
- Powerful digital signal processing (DSP) to ensure superb audio quality; advanced adaptive jitter control and packet loss concealment technology
- Support various vocoders including G.711 (a-law and u-law), G.723.1 (5.3K/6.3K), G.726 (32K), as well as G.729A, and iLBC.
- Support Caller ID/Name display or block, Hold, Call Waiting/Flash, Call Transfer, Call Forward, 3-way conferencing, Call Waiting Caller ID, in-band and out-of-band DTMF, Dial Plans, etc.
- Support fax pass through (for PCMU and PCMA) and T.38 FoIP (Fax over IP)
- Support syslog
- Support volume amplification
- Support configurable Call Progress Tones
- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), Line Echo Cancellation (G.168), and AGC (Automatic Gain Control)
- Support standard encryption and authentication (DIGEST using MD5 and MD5-sess)
- Support for Layer 2 (802.1Q VLAN, 802.1p) and Layer 3 QoS (ToS, DiffServ, MPLS)
- Support automated NAT traversal without manual manipulation of firewall/NAT
- Support device configuration via built-in IVR, Web browser or central configuration file through TFTP or HTTP
- Support firmware upgrade via TFTP or HTTP with encrypted configuration files.
- Ultra compact (wallet size) and lightweight design, great companion for travelers
- Compact, lightweight Universal Power adapter.

4.2   Hardware Specification

The table below lists the hardware specification of HandyTone-496.

Model	**HandyTone-496**
LAN interface	1xRJ45 10Base-T
WAN interface	1xRJ45 10Base-T
FXS telephone port	2xFXS
Button	1
LED	Green and red color
Universal Switching Power Adaptor	Input: 100-240VAC 50-60 Hz Output: +5VDC, 1200mA, **UL** certified
Dimension	70mm (W) 130mm (D) 27mm (H)
Weight	0.6lbs (0.3kg)
Temperature	40 - 130°F 5 – 45°C
Humidity	10% - 90% (non-condensing)
Compliance	FCC/CE/C-Tick

5   Basic Operations

5.1   Get Familiar with Voice Prompt

HandyTone-496 stores a voice prompt menu (Interactive Voice Response or IVR) for quick browsing and simple configuration. Currently, the voice prompt menu and the LED button are designed for FXS port 1 only.

To enter this voice prompt menu, simply pick up the phone and press the button on the HandyTone-496 or pick up the phone and dial "***". The following table shows how to use the voice prompt menu to configure the device.

Menu	Voice Prompt	Options
Main Menu	"Enter a Menu Option"	Enter "*" for the next menu option Enter "#" to return to the main menu Enter 01-06, 47, 86, 99 menu option
01	"DHCP Mode", "Static IP Mode"	Enter "9" to toggle the selection *If user selects "Static IP Mode", user need configure the all IP address information through menu 02 to 05. If user selects "Dynamic IP Mode", the device will retrieve all IP address information from DHCP server automatically when user reboots the device.*
02	"IP Address " + IP address	The current WAN IP address is announced Enter 12 digit new IP address if in Static IP Mode
03	"Subnet " + IP address	Same as menu 02
04	"Gateway " + IP address	Same as menu 02
05	"DNS Server " + IP address	Same as menu 02
06	"TFTP Server " + IP address	Same as menu 02
47	"Direct IP Calling"	When entered, user will be prompted a dial tone, dial a 12 digit IP address to make a direct IP call. (For details, see "4.2.2 Make a Direct IP Call".)
99	"RESET"	Enter "9" to reboot the device; or Enter encoded MAC address to restore factory default setting (For details, see section 8.)
	"Invalid Entry"	Automatically returns to main menu

NOTES:

- *Once the button is pressed, it enters the voice prompt main menu. If the button is pressed again, while it is already in the voice prompt menu, it jumps to "Direct IP Call" option and a dial tone is prompted*
- *"*" shifts down to the next menu option*
- *"#" returns to the main menu*
- *"9" functions as the ENTER key in many cases to confirm an option*
- *All entered digit sequences have known lengths - 2 digits for menu option and 12 digits for IP address. Once all of the digits are collected, the input will be processed.*
- *Key entry can not be deleted but the phone may prompt error once it is detected*

5.2   Make Phone Calls

5.2.1 Calling Phone or Extension Numbers

To make a phone or extension number call:
     a) Dial the number directly and wait for 4 seconds (Default "No Key Entry Timeout"). Or
     b) Dial the number directly, and press # (assuming that "Use # as dial key" is selected in web
       configuration).

Examples:
     To dial another extension on the same proxy, such as 1008, simply pick up the attached phone, dial 1008 and then press the # or wait for 4 seconds.
     To dial a PSTN number such as 6266667890, you might need to enter in some prefix number followed by the phone number. Please check with your VoIP service provider to get the information. If you phone is assigned with a PSTN-like number such as 6265556789, most likely you just follow the rule to dial 16266667890 as if you were calling from a regular analog phone, followed by pressing # or wait for 4 seconds.

5.2.2 Direct IP Calls

Direct IP calling allows two parties, that is, a HandyTone with an analog phone and another VoIP Device, to talk to each other in an ad hoc fashion without a SIP proxy. This kind of VoIP calls can be made between two parties if:

- Both HandyTone ATA and other VoIP Device(i.e., another HandyTone ATA or Budgetone SIP phone or other VoIP unit) have public IP addresses, or
- Both HandyTone ATA and other VoIP Device are on the same LAN using private IP addresses, or
- Both HandyTone ATA and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

To make a direct IP call, first pick up the analog phone or turn on the speakerphone on the analog phone, then access the voice menu prompt by dial "***" or press the button on the HandyTone-496, and dials "47" to access the direct IP call menu. User will hear a voice prompt "Direct IP Calling" and a dial tone. Enter a 12-digit target IP address to make a call. Destination ports can be specified by using "*4" (encoding for ":") followed by the port number.

Examples:

If the target IP address is 192.168.0.160, the dialing convention is
Voice Prompt with option 47, then 192168000160
followed by pressing the "#" key if it is configured as a send key or wait 4 seconds. In this case, the default destination port 5060 is used if no port is specified.

If the target IP address/port is 192.168.1.20:5062, then the dialing convention would be:
Voice Prompt with option 47, then 192168001020*45062** followed by pressing the "#" key if it is configured as a send key or wait for 4 seconds.

5.2.3   Call Hold

While in conversation, pressing the "flash" button on the attached phone will put the remote end on hold. Pressing the "flash" button again will release the previously held party and the bi-directional media will resume.

5.2.4   Call Waiting

If call waiting feature is enabled, while the user is in a conversation, he will hear a special stutter tone if there is another incoming call. User can press the flash button to put the current call party on hold and switch to the other call. Pressing flash button toggles between two active calls.

5.2.5   Call Transfer

5.2.5.1   Blind Transfer

Assuming that call party A and B are in conversation. A wants to *Blind Transfer* B to C:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. Then A dials *87 then dials C's number, and then #(or wait for 4 seconds)
3. A can hang up.

NOTE: *"Enable Call Feature" has to be set to "Yes" in web configuration page.*

A can hold on to the phone and wait for one of the three following behaviors:

- A quick confirmation tone (temporarily using the call waiting indication tone) followed by a dial tone.   This indicates the transfer is successful (transferee has received a 200 OK from transfer target). At this point, A can either hang up or make another call.

- A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
- Busy tone keeps playing. This means we have failed to receive the second NOTIFY from the transferee and decided to time out. Note: this does not indicate the transfer has been successful, nor does it indicate the transfer has failed. When transferee is a client that does not support the second NOTIFY (such as our own earlier firmware), this will be the case. In bad network scenarios, this could also happen, although the transfer may have been completed successfully.

5.2.5.2   Attended Transfer

Assuming that call party A and B are in conversation. A wants to *Attend Transfer* B to C:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone
2. A then dial C's number then # (or wait for 4 seconds).
3. If C answers the call, A and C are in conversation. Then A can hang up to complete transfer.
4. If C does not answer the call, A can press "flash" back to talk to B.

NOTE:

- *When Attended Transfer failed, if A hangs up, the HandTone-496 will ring user A again to remind A that B is still on the call. A can pick up the phone to restore conversation with B.*

5.2.6 3-way Conferencing

HandyTone-496 supports 3-way conference in two styles: star code style or Bellcore style.

5.2.6.1   Star Code Style 3-way Conference

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials *23 then C's number then # (or wait for 4 seconds).
3. If C answers the call, then A press "flash" to bring B, C in the conference.
4. If C does not answer the call, A can press "flash" back to talk to B.

5.2.6.2   Bellcore Style 3-way Conference

Bellcore style 3-way conference is also supported. To do this, user needs to enable "Use Bell-style 3-way Conference" in FXS1 or FXS2 web configuration.

Assuming that call party A and B are in conversation. A wants to bring C in a conference:

1. A presses FLASH (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. A dials C's number then # (or wait for 4 seconds).
3. If C answers the call, then A press "flash" to bring B, C in the conference.
4. If C does not answer the call, A can press "flash" back to talk to B.

5.3   Call Features

Following table shows the call features of HandyTone-496.

Key	Call Features
*23	3-way conference
*87	Blind Transfer
*30	Block Caller ID (for all subsequent calls)
*31	Send Caller ID (for all subsequent calls)
*67	Block Caller ID (per call)
*82	Send Caller ID (per call)
*50	Disable Call Waiting (for all subsequent calls)
*51	Enable Call Waiting (for all subsequent calls)
*70	Disable Call Waiting. (Per Call)
*71	Enable Call Waiting (Per Call)
*72	Unconditional Call Forward. To use this feature, dial "*72" and get the dial tone. Then dial the forward number and "#" for a dial tone, then hang up.
*73	Cancel Unconditional Call Forward To cancel "Unconditional Call Forward", dial "*73" and get the dial tone, then hang up.
*90	Busy Call Forward To use this feature, dial "*90" and get the dial tone. Then dial the forward number and "#" for a dial tone, then hang up.
*91	Cancel Busy Call Forward To cancel "Busy Call Forward", dial "*91" and get the dial tone, then hang up
*92	Delayed Call Forward To use this feature, dial "*92" and get the dial tone. Then dial the forward number and "#" for a dial tone, then hang up.
*93	Cancel Delayed Call Forward

	To cancel this Forward, dial "*93" and get the dial tone, then hang up
Flash/Hook	When in conversation, this action will switch to the new incoming call if there is a call waiting indication. When in conversation without an incoming call, this action will switch to a new channel for a new call.

5.4   Fax

HandyTone-496 supports FAX in two modes: T.38 (Fax over IP) and fax pass through. T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting Fax mode to be T.38. If the service provider does not support T.38, pass-through mode may be used. To send or receive faxes in fax pass through mode, user is recommended to select all the Preferred Codecs to be PCMU/PCMA.

5.5   LED Light Pattern Indication

Following tables show the LED light pattern indication. The LED shows PHONE1 status only.

RED LED always indicates not abnormal status	
DHCP Failed or WAN No Cable	Button flashes every 2 seconds (if DHCP is configured)
HandyTone-496 fails to register	Button flashes every 2 seconds (if SIP server is configured)
Firmware Upgrading	Button flashes every 2 seconds
Device Malfunctions	Red light steady on

GREEN LED mostly indicates normal working status	
Message Waiting Indication	Button flashes every 2 seconds
RINGING	Button flashes at 1/10 second
RINGING INTERVAL	Button flashes every second
In Conversation	Green light steady on

6 Configuration Guide

6.1 Configuring HandyTone-496 WAN IP through Voice Prompt

6.1.1 DHCP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-496 to use DHCP.

6.1.2 STATIC IP Mode

Follow section 5.1 with voice menu option 01 to enable HandyTone-496 to use STATIC IP mode, then use option 02, 03, 04, 05 to set up IP address, Subnet Mask, Gateway and DNS server respectively.

6.1.3 TFTP Server Address

Follow section 5.1 with voice menu option 06 to configure the IP address of the TFTP firmware and configuration file server.

6.2 Configuring HandyTone-496 with Web Browser

HandyTone-496 has an embedded Web server that will respond to HTTP GET/POST requests. It also has embedded HTML pages that allow users to configure the HandyTone-496 through a Web browser such as Microsoft's IE and AOL's Netscape.

6.2.1 Access the Web Configuration Menu

The HandyTone-496 HTML configuration menu can be accessed via LAN or WAN port:

- From the LAN port:

  - Directly connect a computer to the LAN port.
  - Open a command window on the computer
  - Type in "ipconfig /release", the IP address etc. becomes 0.
  - Type in "ipconfig /renew", the computer gets an IP address in 192.168.2.x segment by default
  - Open a web browser, type in the default gateway IP address. You will see the login page of the device.

  http://192.168.2.1

- The WAN port HTML configuration option is disabled by default from factory. To access the HTML configuration menu from the WAN port, first enable the "WAN side HTTP access"

option by accessing the configuration via LAN port. With the WAN side HTTP access enabled, then get the WAN IP address of the HandyTone-496 through section 5.1 with menu option 02. The HandyTone-496's Web Configuration page can be accessed by the following URI via WAN port:

http://HandyTone-IP-Address

where the *HandyTone-IP-Address* is the WAN IP address of the HandyTone-496.

NOTE:

- To type IP address into browser to get into the configuration page, please strip out the leading "0" as the browser will parse in octet. e.g.: if the IP address is: 192.168.001.014, please type in: 192.168.1.14.

6.2.2 End User Configuration

Once this HTTP request is entered and sent from a Web browser, the HandyTone-496 will respond with the following login screen:

Grandstream Device Configuration

Password [   ]

[Login]

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The password is case sensitive and the factory default password for End User and administrator is "123" and "admin" respectively. Only administrator can get access to the "ADVANCED SETTING" configuration page.

NOTE:

If you cannot log into the configuration page by using default password, please check with the VoIP service provider. Most likely the service provider has provisioned the device and configured for you.

After a correct password is entered in the login screen, the embedded Web server inside the HandyTone-496 will respond with the Configuration page, which is explained in details below.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT1 FXS PORT2

End User Password:	(purposely not displayed for security protection)
Web Port:	80 (default for HTTP is 80)

IP Address:

◉ dynamically assigned via DHCP (default) or PPPoE (will attempt PPPoE if DHCP fails and following is non-blank)

DHCP hostname:	
DHCP domain:	
DHCP vendor class ID:	
PPPoE account ID:	
PPPoE password:	
PPPoE Service Name:	
Preferred DNS server:	0 . 0 . 0 . 0

○ statically configured as:

IP Address:	192 . 168 . 1 . 16
Subnet Mask:	255 . 255 . 255 . 0
Default Router:	192 . 168 . 1 . 1
DNS Server 1:	204 . 181 . 101 . 4
DNS Server 2:	0 . 0 . 0 . 0

Time Zone:	GMT-5:00 (US Eastern Time, New York) ▼
Daylight Savings Time:	○ No ◉ Yes Optional Rule: 4,1,7,2,0;10,-1,7,2,0;60

NAT/DHCP Server Information & Configuration:

Device Mode:	◉ NAT Router ○ Bridge
Cloned WAN MAC Addr:	(in hex format)
LAN Subnet Mask:	255.255.255.0 (default is 255.255.255.0)
LAN DHCP Base IP:	192.168.2.1 (base IP for the LAN port, default is 192.168.2.1)

DHCP IP Lease Time:	120	(in units of hours, default is 120 hours or 5 days)							
DMZ IP:									
Port Forwarding:	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	
	WAN port	0	LAN IP		LAN port	0	Protocol	UDP Only	

Update   Cancel   Reboot

End User Password	This contains the password to access the Web Configuration Menu. This field is case sensitive.
Web Port	This is the device's internal HTTP server port. Default is 80.
IP Address	There are 2 modes under which the HandyTone ATA can operate: - If DHCP mode is enabled, then all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The HandyTone ATA will acquire its IP address from the first DHCP server it discovers from the LAN it is connected. To use the PPPoE feature the PPPoE account settings need to be set. The HandyTone will attempt to establish a PPPoE session if any of the PPPoE fields is set. - If Static IP mode is enabled, then the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields will need to be configured. These fields are reset to zero by default.
DHCP hostname	This option specifies the name of the client. This field is optional but may be required by some Internet Service Providers. Default is blank.
DHCP domain	This option specifies the domain name that client should use when resolving hostnames via the Domain Name System. Default is blank.

DHCP vendor class ID	This option is used by clients and servers to exchange vendor-specific information. Default is blank.
Time Zone	This parameter controls how the displayed date/time will be adjusted according to the specified time zone.
Daylight Savings Time	This parameter controls whether the displayed time will be daylight savings time or not. If set to "Yes" and the Optional Rule is empty, then the displayed time will be 1 hour ahead of normal time.
	The "Automatic Daylight Saving Time Rule" shall have the following syntax: start-time;end-time;saving Both start-time and end-time have the same syntax: month,day,weekday,hour,minute month: 1,2,3,..,12 (for Jan, Feb, .., Dec) day: [+|-]1,2,3,..,31 weekday: 1, 2, 3, .., 7 (for Mon, Tue, .., Sun), or 0 which means the daylight saving rule is not based on week days but based on the day of the month. hour: hour (0-23), minute: minute (0-59)
	If "weekday" is 0, it means the date to start or end daylight saving is at exactly the given date. In that case, the "day" value must not be negative. If "weekday" is not zero and "day" is positive, then the daylight saving starts on the first "day"th iteration of the weekday (1st Sunday, 3rd Tuesday etc). If "weekday" us not zero and "day" is negative, then the daylight saving starts on the last "day"th iteration of the weekday (last Sunday, 3rd last Tuesday etc).
	The saving is in the unit of minutes. The saving time may also be preceded by a negative (-) sign if subtraction is desired instead of addition.
	The default value for "Automatic Daylight Saving Time Rule" shall be set to "04,01,7,02,00;10,-1,7,02,00;60" which is the rule for US.
	Examples US/Canada where daylight saving time is applicable: 04,01,7,02,00;10,-1,7,02,00;60 This means the daylight saving time starts from the first Sunday of April at 2AM and ends the last Sunday of October at 2AM. The saving is 60 minutes (1hour).
PPPoE account ID	PPPoE username. Fill this field if your ISP requires you to use a PPPoE (Point to Point Protocol over Ethernet) connection.
PPPoE password	PPPoE account password.

PPPoE Service Name	This field is optional. If your ISP uses a service name for the PPPoE connection, enter the service name here. Default is blank.
Device Mode	To use the device as a router or a bridge.
Cloned WAN MAC Address:	Allow the user to set a specific MAC address. Set in Hex format
LAN Subnet Mask	Sets the LAN subnet mask. Default value is 255.255.255.0
LAN DHCP Base IP:	Base IP for the LAN port which functions as a Gateway for the subnet. Default value is 192.168.2.1
DHCP IP Lease Time:	Value is set in units of hours. Default value is 120hr (5 Days.) The time IP address are assigned to the LAN clients
DMZ IP:	Forward all WAN IP traffic to a specific IP address if no matching port is used by HandyTone-496 itself or in the defined port forwarding.
Port Forwarding:	Allow users to forward a matching (TCP/UDP) port to a specific LAN IP address with a specific (TCP/UDP) port.

In addition to the Basic Settings configuration page, end users also have access to the device Status page. The following is a snap shot of the device Status page. Details will be explained next.

Grandstream Device Configuration

STATUS   BASIC SETTINGS   ADVANCED SETTINGS   FXS PORT1 FXS PORT2

MAC Address:	00.0B.82.03.C8.AF
WAN IP Address:	192.168.1.187
Product Model:	HT496
Software Version:	Program-- 1.0.3.44 Bootloader-- 1.0.8.11 HTML-- 1.0.3.44 VOC-- 1.0.0.10
System Up Time:	2 day(s) 0 hour(s) 9 minute(s)
Registered:	Yes
PPPoE Link Up:	disabled
NAT:	detected NAT type is full cone

MAC Address	The device ID, in HEX format. This is a very important ID for ISP troubleshooting.
WAN IP Address	This field shows WAN port IP address.
Product Model	This field contains the product model info.
Software Version	Program: This is the main software release, its number is always used for firmware upgrade. Bootloader: This is normally not changed. HTML: This is the web user interface, normally not changed. VOC: This is the codec program, normally not changed.
System Up Time	This field indicates how long the device has been up since the last reboot.
Registered	This field indicates whether the device is registered to the SIP server.
PPPoE Link Up	This field shows whether the PPPoE connection is enabled or not.
NAT	This field shows what kind NAT the HandyTone is connected to via its WAN port. It is based on STUN protocol.

6.2.3 Advanced User Configuration

To login to the Advanced User Configuration page, please follow the instructions in section 6.2.1 to get to the following login page. The password is case sensitive and the factory default password for Advanced User is "admin".

Grandstream Device Configuration

Password [   ]

Login

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Advanced User configuration includes not only the end user configuration, but also advanced configuration such as SIP configuration, Codec selection, NAT Traversal Setting and other miscellaneous configuration.   Following is a snap shot of the advanced configuration page.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT1 FXS PORT2

Admin Password: [   ] (purposely not displayed for security protection)

Home NPA: [   ]

Layer 3 QoS: [48] (Diff-Serv or Precedence value)

Layer 2 QoS: 802.1Q/VLAN Tag [0]   802.1p priority value [0] (0-7)

STUN server is : [stun.mycompany.com] (URI or IP:port)

keep-alive interval: [20] (in seconds, default 20 seconds)

Use NAT IP: [   ] (used in SIP/SDP message if specified)

Firmware Upgrade and Provisioning:

Upgrade Via ⊙ TFTP   ○ HTTP

Firmware Server Path: [192.168.1.193]

Config Server Path: [192.168.1.193]

Firmware File Prefix: [   ]   Firmware File Postfix: [   ]

Config File Prefix: [   ]   Config File Postfix: [   ]

Automatic Upgrade:
⊙ No   ○ Yes, check for upgrade every [10080] minutes (default 7 days)

⊙ Always Check for New Firmware

○ Check New Firmware only when F/W pre/suffix changes

Firmware Key: [   ] (in Hexadecimal Representation)

Onhook Threshold: [800 ms ▼]

FXS Impedance: [600 Ohm (North America) ▼]

Caller ID Scheme: [Bellcore (North America) ▼]

Onhook Voltage: [36V ▼]

Polarity Reversal: ⊙ No   ○ Yes (reverse polarity upon call establishment and termination)

NTP Server: [time.nist.gov] (URI or IP address)

Reply to ICMP on WAN port: ⊙ No   ○ Yes (Unit will not respond to PING from WAN side if set to No)

WAN side http access:	☐ No ☑ Yes (WAN side access to http server will be rejected if set to No)
Syslog Server:	
Syslog Level:	NONE ▼

[Update]   [Cancel]   [Reboot]

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Two FXS SIP accounts each has its own configuration page. Their configurations are identical. The following is a screen shot of FXS Port 1 Account settings.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS **FXS PORT1** FXS PORT2

SIP Server:	sip.mycompany.com	(e.g., sip.mycompany.com, or IP address)
Outbound Proxy:		(e.g., proxy.myprovider.com, or IP address, if any)
SIP User ID:	3123320	(the user part of an SIP address)
Authenticate ID:	3123320	(can be identical to or different from **SIP User ID**)
Authenticate Password:		(purposely not displayed for security protection)
Name:	John Doe	(optional, e.g., John Doe)

Use DNS SRV:	☑ No ☐ Yes
User ID is phone number:	☑ No ☐ Yes
SIP Registration:	☐ No ☑ Yes
Unregister On Reboot:	☑ No ☐ Yes
Register Expiration:	60 (in minutes. default 1 hour, max 45 days)
local SIP port:	5060 (default 5060)
local RTP port:	5004 (1024-65535, default 5004)
Use random port:	☑ No ☐ Yes
DTMF Payload Type:	101
Send DTMF:	☑ in-audio ☐ via RTP (RFC2833) ☐ via SIP INFO

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Send Flash Event:	⦿ No ◯ Yes (Flash will be sent as a DTMF event if set to Yes)	
Enable Call Features:	◯ No ⦿ Yes (if Yes, Call Forwarding & Call-Waiting-Disable are supported locally)	
Use Bell-style 3-way Conference:	⦿ No ◯ Yes (if Yes, *23 will be disabled)	
Offhook Auto-Dial:	[ ] (User ID/extension to dial automatically when offhook)	
Proxy-Require:	[ ]	
Disable Call-Waiting:	⦿ No ◯ Yes	
NAT Traversal (STUN):	◯ No ⦿ Yes	
No Key Entry Timeout:	[4] (in seconds, default is 4 seconds)	

Preferred Vocoder: (in listed order)	choice 1:	current setting is " PCMU" ▼
	choice 2:	current setting is " PCMA" ▼
	choice 3:	current setting is " G723" ▼
	choice 4:	current setting is " G729" ▼
	choice 5:	current setting is " G726-32" ▼
	choice 6:	current setting is " iLBC" ▼
Voice Frames per TX:	[2] (up to 10/20/32/64 for G711/G726/G723/other codecs respectively)	
G723 rate:	⦿ 6.3kbps encoding rate ◯ 5.3kbps encoding rate	
iLBC frame size:	⦿ 20ms ◯ 30ms	
iLBC payload type:	[97] (between 96 and 127, default is 97)	
Silence Suppression:	⦿ No ◯ Yes	
Fax Mode:	⦿ T.38 (Auto Detect) ◯ Pass-Through	
Early Dial:	⦿ No ◯ Yes (use "Yes" only if proxy supports 484 response)	
Dial Plan Prefix:	[ ] (this prefix string is added to each dialed number)	
Use # as Dial Key:	◯ No ⦿ Yes (if set to Yes, "#" will function as the "(Re-)Dial" key)	
SUBSCRIBE for MWI:	⦿ No, do not send SUBSCRIBE for Audible Ringing Indication ◯ Yes, send periodical SUBSCRIBE for Audible Ringing Indication	
Send Anonymous:	⦿ No ◯ Yes (caller ID will be blocked if set to Yes)	
Lock keypad update:	⦿ No ◯ Yes (configuration update via keypad is disabled if set to Yes)	

	Frequency1(Hz)	Frequency2(Hz)	ON (x10ms)	OFF (x10ms)	Alert-info
Special Feature:	Standard				
	350	440	0	0	Dial Tone
	350	440	10	10	Recall Dial Tone
	350	440	10	10	Message Waiting
	350	440	10	10	Confirmation
	440	480	200	400	Audible Ringing
	480	620	50	50	Busy Tone
	480	620	25	25	Reorder Tone
	1400	2600	10	10	Receiver offhook
	440	440	100	100	Ring1
	440	440	100	100	Ring2
Distinct Ring Tones:	440	440	100	100	Ring3
	440	440	100	100	Ring4
	440	440	100	100	Ring5
	440	440	100	100	Ring6
	20	20	80	40	Ring7
	20	20	40	20	Ring8
	20	20	30	20	Ring9
	20	20	50	800	Ring10
	20	20	200	400	Ring11
	20	20	200	400	Ring12

Volume Amplification: TX [0dB default]   RX [0dB default]

[Update]   [Cancel]   [Reboot]

Admin Password	This contains the password to access the Advanced Web Configuration page. This field is case sensitive.
Home NPA	Local area code for North American Dial Plan.
Layer 3 QoS	This field defines the layer 3 QoS parameter which can be the value used for IP Precedence or Diff-Serv or MPLS. Default value is 48.

Layer 2 QoS	This setting includes two fields. The 802.1Q/VLAN Tag contains the value used for layer 2 VLAN tag. Default setting is blank. And 802.1p priority value contains the value of the priority value.
STUN Server	STUN server's IP or FQDN
Keep-alive interval	This parameter specifies how often the HandyTone ATA sends a blank UDP packet to the SIP server in order to keep the "hole" on the NAT open.
Use NAT IP:	NAT IP address is used in SIP/SDP message. Default is blank.
Firmware Upgrade and Provisioning	Default method is HTTP. Firmware upgrade may take up to 10 minutes depending on network environment. Do not interrupt the firmware upgrading process.
Firmware Server Path	IP address or domain name of firmware server.
Config Server Path	IP address or domain name of configuration server.
Firmware File Prefix	Default is blank. If configured, HandyTone-496 will request the firmware file with the prefix. This setting is useful for ITSPs. End user should keep it blank.
Firmware File Postfix	Default is blank. End user should keep it blank.
Config File Prefix	Default is blank. End user should keep it blank.
Config File Postfix	Default is blank. End user should keep it blank.
Automatic Upgrade	Default is "Yes".
Firmware Key	For firmware encryption. It should be 32 digit in Hexadecimal Representation. End user should keep it blank.
Onhook Threshold	The amount of time the hookflash is pressed that will cause the device to onhook. Default is 800ms.
FXS Impedance	Selects the impedance of the analog telephone connected to the Phone port.
Caller ID Scheme	• Bellcore (North America) • CID-Canada • DTMF-Brazil • DTMF-Sewden • DTMF (Denmark) • ETSI-DTMF (Finland, Sweden) • ETSI-FSK (France, Germany, Norway, Taiwan, UK-CCA)
Onhook Voltage	Select the onhook voltage to suit different area or PBX.

Polarity Reversal	Default is No. If set to Yes, polarity will be reversed upon call establishment and termination.
NTP server	This parameter defines the URI or IP address of the NTP server which is used by the HandyTone ATA to display the current date/time.
Reply to ICMP on WAN port	When this parameter is set to "No", the HandyTone ATA will not respond to PING from WAN side.
WAN side http access	If this parameter is set to "No", the HTML configuration update via WAN port is disabled.
Syslog Server	The IP address or URL of System log server. This feature is especially useful for ITSP (Internet Telephone Service Provider)
Syslog Level	Select the ATA to report the log level. Default is NONE. The level is one of DEBUG, INFO, WARNING or ERROR. Syslog messages are sent based on the following events: • product model/version on boot up (INFO level) • NAT related info (INFO level) • sent or received SIP message (DEBUG level) • SIP message summary (INFO level) • inbound and outbound calls (INFO level) • registration status change (INFO level) • negotiated codec (INFO level) • Ethernet link up (INFO level) • SLIC chip exception (WARNING and ERROR levels) • memory exception (ERROR level) The Syslog uses USER facility. In addition to standard Syslog payload, it contains the following components: GS_LOG: [device MAC address][error code] error message Here is an example: May 19 02:40:38 192.168.1.14 GS_LOG: [00:0b:82:00:a1:be][000] Ethernet link is up

Individual Account Settings

SIP Server	This field contains the URI string or the IP address (and port, if different from 5060) of the SIP proxy server. e.g., the following are some valid examples: sip.my-voip-provider.com, or sip:my-company-sip-server.com, or 192.168.1.200:5066
Outbound Proxy	This field contains the URI string or the IP address (and port, if different from 5060) of the outbound proxy. If there is no outbound proxy, this field **SHOULD** be left blank. If not blank, all outgoing requests will be sent to this outbound proxy.
SIP User ID	This field contains the user part of the SIP address for this phone. e.g., if the SIP address is: sip:my_user_id@my_provider.com, then the SIP User ID is: my_user_id. Please do NOT include the preceding "sip:" scheme or the host portion of the SIP address in this field.
Authenticate ID	SIP service subscriber's Authenticate ID. Can be identical to or different from SIP User ID.
Authenticate Password	SIP service subscriber's account password.
Name	SIP service subscriber's name which will be used for Caller ID display.
Use DNS SRV:	Default is No. If set to Yes the client will use DNS SRV for server lookup.
User ID is Phone Number	If the HandyTone ATA has an assigned PSTN telephone number, this field should be set to "Yes". Otherwise, set it to "No". If "Yes" is set, a "user=phone" parameter will be attached to the "From" header in SIP request.
SIP Registration	This parameter controls whether the HandyTone ATA needs to send REGISTER messages to the proxy server. The default setting is "Yes".
Unregister on Reboot	Default is No. If set to yes, the SIP user's registration information will be cleared on reboot.
Register Expiration	This parameter allows the user to specify the time frequency (in minutes) the HandyTone ATA refreshes its registration with the specified registrar. The default interval is 60 minutes (or 1 hour). The maximum interval is 65535 minutes (about 45 days).
Local SIP port	This parameter defines the local SIP port the HandyTone ATA will listen and transmit. The default value for FXS port 1 is 5060. The default value for FXS port 2 is 5062.
Local RTP port	This parameter defines the local RTP-RTCP port pair the HandyTone ATA will listen and transmit. It is the base RTP port for channel 0. When configured, channel 0 will use this port _value for RTP and the port_value+1 for its RTCP; channel 1 will use port_value+2 for RTP and port_value+3 for its RTCP. The default value for FXS port 1 is 5004. The default value for FXS port 2 is 5008.

Use Random Port	This parameter, when set to Yes, will force random generation of both the local SIP and RTP ports. This is usually necessary when multiple HandyTone ATAs are behind the same NAT.
DTMF Payload Type	This parameter sets the payload type for DTMF using RFC2833.
Send DTMF	This parameter specifies the mechanism to transmit DTMF digit. There are 3 modes supported: in audio which means DTMF is combined in audio signal (not very reliable with low-bit-rate codec), via RTP (RFC2833), or via SIP INFO. Multiple selections of DTMF method are supported.
Send Flash Event	Default is NO. If set to yes, flash will be sent as DTMF event.
Enable Call Features	Default is No. If set to Yes, Call Forwarding & Do-Not-Disturb are supported locally
Use Bell-style 3-way Conference	If this parameter is set to "Yes", user will be able to make Bellcore style 3-way conference. *23 will be disabled.
Offhook Auto-Dial	This parameter allows users to configure a User ID or extension number to be automatically dialed upon offhook. Please note that only the user part of a SIP address needs to be entered here. The HandyTone ATA will automatically append the "@" and the host portion of the corresponding SIP address.
Proxy-Require	SIP Extension to notify SIP server that the unit is behind the NAT/Firewall.
Disable Call Waiting	Default is No.
NAT Traversal (STUN)	This parameter defines whether the HandyTone ATA NAT traversal mechanism will be activated or not. If activated (by choosing "Yes") and a STUN server is also specified, then the HandyTone ATA will behave according to the STUN client specification. Under this mode, the embedded STUN client inside the HandyTone ATA will attempt to detect if and what type of firewall/NAT it is sitting behind through communication with the specified STUN server. If the detected NAT is a Full Cone, Restricted Cone, or a Port-Restricted Cone, the HandyTone ATA will attempt to use its mapped public IP address and port in all its SIP and SDP messages. If the NAT Traversal field is set to "Yes" with no specified STUN server, the HandyTone ATA will periodically (every 20 seconds or so) send a blank UDP packet (with no payload data) to the SIP server to keep the "hole" on the NAT open.
No Key Entry Timeout	Default is 4 seconds.

Preferred Vocoder	The HandyTone ATA supports up to 6 different Vocoder types including G.711 A-/U-law , G.723.1, G.726, G.729A/B, iLBC. Depending on the product model, some of these Vocoders may not be provided in standard release. Users can configure Vocoders in a preference list that will be included with the same preference order in SDP message. The first Vocoder in this list can be entered by choosing the appropriate option in "Choice 1". Similarly, the last Vocoder in this list can be entered by choosing the appropriate option in "Choice 6".
G723 Rate:	This defines the encoding rate for G723 vocoder. By default, 6.3kbps rate is chosen.
iLBC frame size:	This sets the iLBC frame size in 20ms or 30ms
iLBC payload type:	This defines payload time for iLBC. Default value is 97. The valid range is between 96 and 127.
Voice Frames per TX	This field contains the number of voice frames to be transmitted in a single packet. When setting this value, the user should be aware of the requested packet time (used in SDP message) as a result of configuring this parameter. This parameter is associated with the first vocoder in the above vocoder Preference List or the actual used payload type negotiated between the 2 conversation parties at run time. e.g., if the first vocoder is configured as G723 and the "Voice Frames per TX" is set to be 2, then the "ptime" value in the SDP message of an INVITE request will be 60ms because each G723 voice frame contains 30ms of audio. Similarly, if this field is set to be 2 and if the first vocoder chosen is G729 or G711 or G726, then the "ptime" value in the SDP message of an INVITE request will be 20ms. If the configured voice frames per TX exceeds the maximum allowed value, the HandyTone ATA will use and save the maximum allowed value for the corresponding first vocoder choice. The maximum value for PCM is 10(x10ms) frames; for G726, it is 20 (x10ms) frames; for G723, it is 32 (x30ms) frames; for G729/G728, 64 (x10ms) and 64 (x2.5ms) frames respectively.
Silence Suppression	This controls the silence suppression/VAD feature of G723. If set to "Yes", when a silence is detected, small quantity of VAD packets (instead of audio packets) will be sent during the period of no talking. If set to "No", this feature is disabled.
Fax Mode	T.38 (Auto Detect) FoIP by default, or Pass-Through (must use codec PCMU/PCMA)

Early Dial	This parameter controls whether the phone will attempt to send an early INVITE each time a key is pressed when a user dials a number. If set to "Yes", an INVITE is sent using the dial-number collected thus far; Otherwise, no INVITE is sent until the "(Re-)Dial" button is pressed or after about 5 seconds have elapsed if the user forgets to press the "(Re-)Dial" button. The "Yes" option should be used ONLY if there is a SIP proxy configured and the proxy server supports 484 Incomplete Address response. Otherwise, the call will most likely be rejected by the proxy (with a 404 Not Found error). Please note that this feature is NOT designed to work with and should NOT be enabled for direct IP-to-IP calling.
Dial Plan Prefix	Sets the prefix added to each dialed number
Use # as Send Key	This parameter allows users to configure the "#" key to be used as the "Send" (or "Dial") key. If set to "Yes", pressing this key will immediately trigger the sending of dialed string collected so far. In this case, this key is essentially equivalent to the "(Re)Dial" key. If set to "No", this "#" key will then be included as part of the dial string to be sent out.
Subscribe for MWI:	Default is No. When set to "Yes" a SUBSCRIBE for Message Waiting Indication will be sent periodically.
Send Anonymous	If this parameter is set to "Yes", the "From" header in outgoing INVITE message will be set to anonymous, essentially blocking the Caller ID from displaying.
Lock keypad update	If this parameter is set to "Yes", the configuration update via keypad is disabled.
Special Feature	Default is Standard. Choose the selection to meet some special requirements from Soft Switch vendors like Lucent FS5000 Simple Endpoint, CBCOM, etc.
Distinct Ring Tones	Using these settings, user can configure ring or tone frequencies according to their preference. By default they are set to North American frequencies. Frequencies should be configured with known values to avoid uncomfortable high pitch sounds. ON is the period of ringing ("On time" in 'ms') while OFF is the period of silence. In order to set a continuous ring, OFF should be zero. Otherwise it will ring ON ms and a pause of OFF ms and then repeat the pattern. Ring 1 - 6 are reserved for distinct call waiting tones through SIP Alert-Info. Ring 7 – 12 are distinct ring tones through SIP Alert-Info. Both client and Server need to support this to be a useful feature. Currently, distinct ring tone feature is only supported under Lucent Special Feature.
Volume Amplification	Handset volume adjustment. RX is for receiving volume, TX is for transmission volume. Default values are 0dB for both parameters. +6dB generates the highest volume and -6dB generates the lowest volume.

6.2.4 Saving the Configuration Changes

Once a change is made, users should click on the "Update" button in the Configuration page. The HandyTone-496 will then display the following screen to confirm that the changes have been saved.

Grandstream Device Configuration

STATUS BASIC SETTINGS ADVANCED SETTINGS FXS PORT1 FXS PORT2

Your configuration changes have been saved.
They will take effect on next reboot.

Reboot

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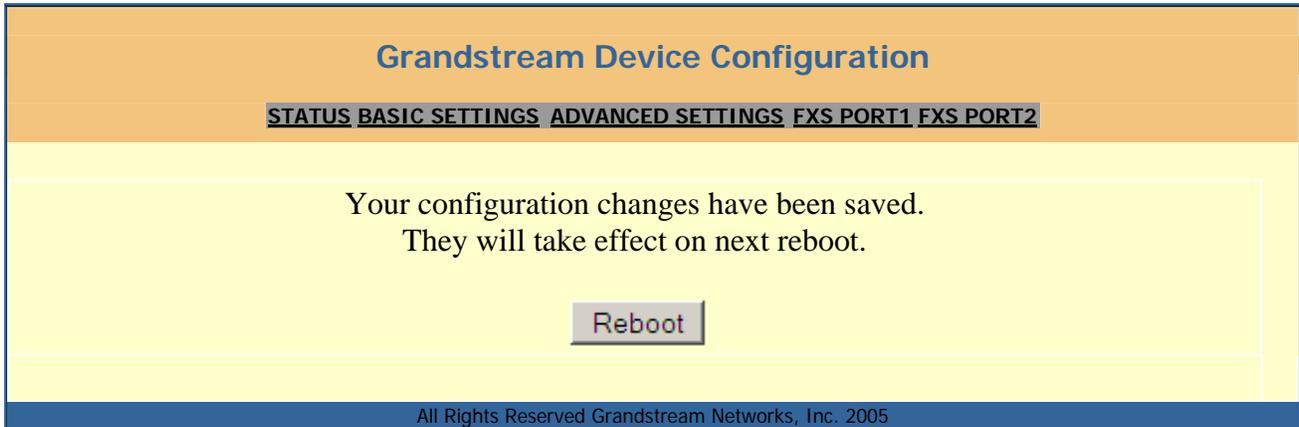
Users are recommended to power cycle the HandyTone-496 after seeing the above message.

6.2.5 Rebooting the HandyTone-496 from Remote

The administrator of the HandyTone-496 can remotely reboot the HandyTone ATA by clicking on the "Reboot" button at the bottom of the configuration page. Once done, the following screen will be displayed to indicate that rebooting is underway.

Grandstream Device Configuration

The device is rebooting now...
You may relogin by clicking on the link below in 30 seconds.
Click to relogin

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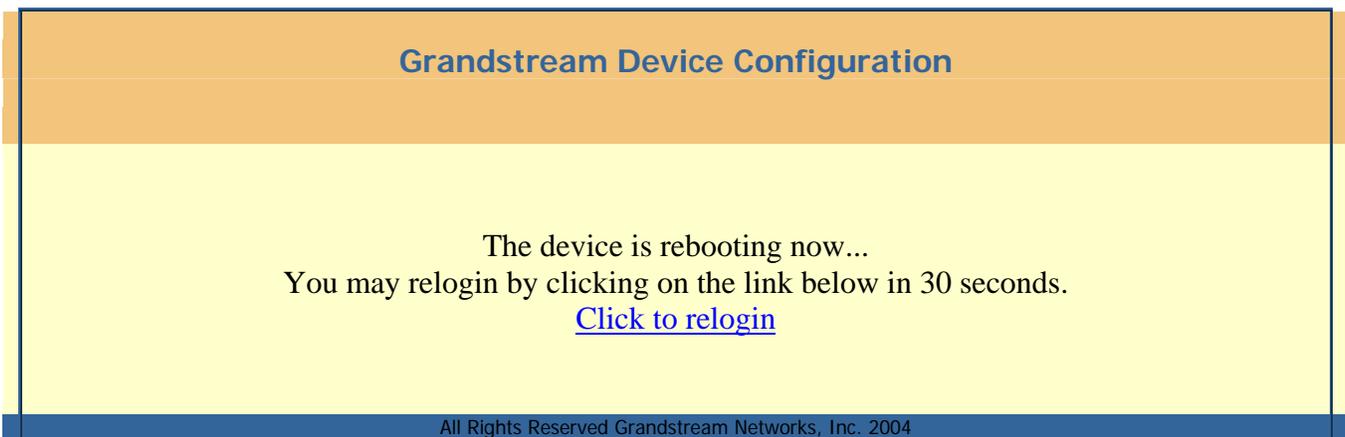
At this point, the user can relogin to the HandyTone-496 after waiting for about 30 seconds.

6.3   Configuration through a Central Server

Grandstream HandyTone ATAs can be automatically configured from a central provisioning system.

When HandyTone ATA boot up, it will send TFTP or HTTP request to download configuration file, "cfg000b82xxxxxx", where "000b82xxxxxx" is the MAC address of the HandyTone ATA.

The configuration files can be downloaded via TFTP or HTTP from the central server. A service provider or an enterprise with large deployment of HandyTone ATA can easily manage the configuration and service provisioning of individual devices remotely from a central server.

Grandstream provides a licensed provisioning system called GAPS that can be used to support automated configuration of HandyTone ATA. GAPS (Grandstream Automated Provisioning System) uses enhanced (NAT friendly) TFTP or HTTP (thus no NAT issues) and other communication protocols to communicate with each individual HandyTone ATA for firmware upgrade, remote reboot, etc.

Grandstream provide GAPS (Grandstream Automated Provisioning System) service to VoIP service providers. It could be either simple redirection or with certain special provisioning settings. Initially upon booting up, Grandstream devices by default point to Grandstream provisioning server GAPS, based on the unique MAC address of each device, GAPS provision the devices with redirection settings so that they will be redirected to customer's TFTP or HTTP server for further provisioning. Grandstream also provide GAPSLite software package which contains our NAT friendly TFTP server and a configuration tool to facilitate the task of generating device configuration files.

The GAPSLite configuration tool is now free to end users. The tool and configuration template are available for download from http://www.grandstream.com/DOWNLOAD/Configuration_Tool/.

For details on how GAPS works, please contact Grandstream and refer to the documentation of GAPS product provided.

7   Software Upgrade

Software upgrade can be done via either TFTP or HTTP. The corresponding configuration settings are in the ADVANCED SETTINGS configuration page.

7.1   Firmware Upgrade through TFTP/HTTP

To upgrade via TFTP or HTTP, the "Firmware Upgrade and Provisioning upgrade via" field needs to be set to TFTP or HTTP, respectively. "Firmware Server Path" needs to be set to a valid URL of a TFTP or HTTP server, server name can be in either FQDN or IP address format. Here are examples of some valid URL.

       e.g. firmware.mycompany.com:6688/Grandstream/1.0.3.44
       e.g. 168.75.215.190

NOTES:

- TFTP server in IP address format can be configured via IVR. Please refer to section 6.1.3 for instructions. If TFTP server is in FQDN format, it must be set via web configuration interface.
- Once a "Firmware Server Path" is set, user needs to update the settings and reboot the device. If the configured firmware server is found and a new code image is available, the HandyTone ATA will attempt to retrieve the new image files by downloading them into the HandyTone ATA's SRAM. During this stage, the HandyTone ATA's LEDs will blink until the checking/downloading process is completed. Upon verification of checksum, the new code image will then be saved into the Flash. If TFTP/HTTP fails for any reason (e.g., TFTP/HTTP server is not responding, there are no code image files available for upgrade, or checksum test fails, etc), the HandyTone ATA will stop the TFTP/HTTP process and simply boot using the existing code image in the flash.
- Firmware upgrade may take as long as 1 to 20 minutes over Internet, or just 20+ seconds if it is performed on a LAN. It is recommended to conduct firmware upgrade in a controlled LAN environment if possible. For users who do not have a local firmware upgrade server, Grandstream provides a NAT-friendly TFTP server on the public Internet for firmware upgrade. Please check the Services section of Grandstream's Web site to obtain our public TFTP server's IP address.
- Alternatively, user can download a free TFTP or HTTP server and conduct local firmware upgrade. A free windows version TFTP server is available for download from http://support.solarwinds.net/updates/New-customerFree.cfm. Our latest official release can be downloaded from http://www.grandstream.com/y-firmware.htm. Unzip the file and put all of them under the root directory of the TFTP server. Put the PC running the TFTP server and the HandyTone ATA in the same LAN segment. Please go to File -> Configure -> Security to change the TFTP server's default setting from "Receive Only" to "Transmit Only" for the firmware upgrade. Start the TFTP server, in the HandyTone ATA's web configuration page, configure the Firmware Server Path with the IP address of the PC, update the change and reboot the unit. Please be advised that our client will pull out firmware from the WAN side, if the TFTP server is connected to the device's LAN port, the firmware upgrade will not work by design.

7.2   Configuration File Download

Grandstream SIP Device can be configured via Web Interface as well as via Configuration File through TFTP or HTTP. "Config Server Path" is the TFTP or HTTP server path for configuration file. It needs to be set to a valid URL, either in FQDN or IP address format. The "Config Server Path" can be same or different from the "Firmware Server Path".

A configuration parameter is associated with each particular field in the web configuration page. A parameter consists of a Capital letter P and 2 to 3 (Could be extended to 4 in the future) digit numeric numbers. i.e., P2 is associated with "Admin Password" in the ADVANCED SETTINGS page. For a detailed parameter list, please refer to the corresponding firmware release configuration template.

When Grandstream Device boots up or reboots, it will issue request for configuration file named "cfgxxxxxxxxxxxx", where "xxxxxxxxxxxx" is the MAC address of the device, i.e., "cfg000b820102ab". The configuration file name should be in lower cases.

7.3   Firmware and Configuration File Prefix and Postfix

Starting from firmware version 1.0.3.18 for HandyTone-496, adding prefix and postfix for both firmware and configuration file is supported.

Firmware Prefix and Postfix allows device to download the firmware name with the matching Prefix and Postfix. This makes it the possible to store ALL of the firmware with different version in one single directory. Similarly, Config File Prefix and Postfix allows device to download the configuration file with the matching Prefix and Postfix. Thus multiple configuration files for the same device can be stored in one directory.

In addition, when the field "Check New Firmware only when F/W pre/suffix changes" is set to "Yes", the device will only issue firmware upgrade request if there are changes in the firmware Prefix or Postfix.

7.4   Managing Firmware and Configuration File Download

When "Automatic Upgrade" is set to "Yes", Service Provider can use P193 (Auto Check Interval, in minutes, default and minimum is 60 minutes) to have the devices periodically check with either Firmware Server or Config Server, whenever they are defined. This allows the device periodically check if there are any new changes need to be taken on a scheduled time. By defining different intervals in P193 for different devices, Server Provider can spread the Firmware or Configuration File download in minutes to reduce the Firmware or Provisioning Server load at any given time.

8   Restore Factory Default Setting

Warning !!!

Restore the Factory Default Setting will DELETE all configuration information of the device. Please backup or print out all the settings before you approach to following steps. Grandstream will not take any responsibility if you lose all the parameters of setting and cannot connect to your service provider.

Please disconnect network cable and power cycle the unit before trying to reset the unit to factory default. The steps are as follows:

Step 1:
        Find the MAC address of the device. It is a 12 digits HEX number located on the bottom of the unit.

Step 2:
        Encode the MAC address. Please use the following mapping:

        0-9: 0-9
        A: 22
        B: 222
        C: 2222
        D: 33
        E: 333
        F: 3333

        For example, if the MAC address is 000b8200e395, it should be encoded as
        "0002228200333395".

Step 3:
        To perform factory reset:

        a. Press "***" or the LED button for voice prompt.
        b. Enter "99" and get the voice prompt "Reset".
        c. Enter the encoded MAC address of the device.
        d. Wait for 15 seconds.

        The device will reboot automatically and restore to factory default setting.

NOTE:

- *Please be aware by default the HandyTone-496 WAN side HTTP access is disabled. After a factory reset, the device's web configuration page can be accessed only from its LAN port, please refer to instructions in section 6.2.1 for details.*

9   Glossary of Terms

ADSL

Asymmetric Digital Subscriber Line: Modems attached to twisted pair copper wiring that transmit from 1.5 Mbps to 9 Mbps downstream (to the subscriber) and from 16 kbps to 800 kbps upstream, depending on line distance.

AGC

Automatic Gain Control, is an electronic system found in many types of devices. Its purpose is to control the gain of a system in order to maintain some measure of performance over a changing range of real world conditions.

ARP

Address Resolution Protocol is a protocol used by the Internet Protocol (IP) [RFC826], pecifically IPv4, to map IP network addresses to the hardware addresses used by a data link protocol. The protocol operates below the network layer as a part of the interface between the OSI network and OSI link layer. It is used when IPv4 is used over Ethernet

ATA

Analogue Telephone Adapter. Covert analogue telephone to be used in data network for VoIP, like Grandstream HT series products.

CODEC

Abbreviation for Coder-Decoder. It's an analog-to-digital (A/D) and digital-to-analog (D/A) converter for translating the signals from the outside world to digital, and back again.

CNG

Comfort Noise Generator, geneate artificial background noise used in radio and wireless communications to fill the silent time in a transmission resulting from voice activity detection.

DATAGRAM

A data packet carrying its own address information so it can be independently routed from its source to the destination computer

DECIMATE

To discard portions of a signal in order to reduce the amount of information to be encoded or compressed. Lossy compression algorithms ordinarily decimate while subsampling.

DECT

Digital Enhanced Cordless Telecommunications: A standard developed by the European Telecommunication Standard Institute from 1988, governing pan-European digital mobile telephony. DECT covers wireless PBXs, telepoint, residential cordless telephones, wireless access to the public switched telephone network, Closed User Groups (CUGs), Local Area Networks, and wireless local loop. The DECT Common Interface radio standard is a multicarrier time division multiple access, time division duplex (MC-TDMA-TDD) radio transmission technique using ten radio frequency channels from 1880 to 1930 MHz, each

divided into 24 time slots of 10ms, and twelve full-duplex accesses per carrier, for a total of 120 possible combinations. A DECT base station (an RFP, Radio Fixed Part) can transmit all 12 possible accesses (time slots) simultaneously by using different frequencies or using only one frequency. All signaling information is transmitted from the RFP within a multiframe (16 frames). Voice signals are digitally encoded into a 32 kbit/s signal using Adaptive Differential Pulse Code Modulation.

DNS

Short for Domain Name System (or Service or Server), an Internet service that translates *domain names* into IP addresses

DID

Direct Inward Dialing

Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant.

DSP

Digital Signal Processing. Using computers to process signals such as sound, video, and other analog signals which have been converted to digital form.

Digital Signal Processor. A specialized CPU used for digital signal processing.

Grandstream products all have DSP chips built inside.

DTMF

Dual Tone Multi Frequency

The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #).

FQDN

Fully Qualified Domain Name

A FQDN consists of a host and domain name, including top-level domain. For example, www.grandstream.com is a fully qualified domain name. www is the host, grandstream is the second-level domain, and.com is the top level domain.

FXO

Foreign eXchange Office

An FXO device can be an analog phone, answering machine, fax, or anything that handles a call from the telephone company like AT&T. They should also operate the same way when connected to an FXS interface.

An FXO interface will accept calls from FXS or PSTN interfaces. All countries and regions have their own standards.

FXO is complimentary to FXS (and the PSTN).

FXS

Foreign eXchange Station

An FXS device has hardware to generate the ring signal to the FXO extension (usually an analog phone).

An FXS device will allow any FXO device to operate as if it were connected to the phone company. This makes your PBX the POTS+PSTN for the phone.

The FXS Interface connects to FXO devices (by an FXO interface, of course).

DHCP

The Dynamic Host Configuration Protocol (DHCP) is an Internet protocol for automating the configuration of computers that use TCP/IP. DHCP can be used to automatically assign IP addresses, to deliver TCP/IP stack configuration parameters such as the subnet mask and default router, and to provide other configuration information such as the addresses for printer, time and news servers.

ECHO CANCELLATION

Echo Cancellation is used in telephony to describe the process of removing echo from a voice communication in order to improve voice quality on a telephone call. In addition to improving quality, this process improves bandwidth savings achieved through silence suppression by preventing echo from traveling across a network.

There are two types of echo of relevance in telephony: acoustic echo and hybrid echo. Speech compression techniques and digital processing delay often contribute to echo generation in telephone networks.

H.323

A suite of standards for multimedia conferences on traditional packet-switched networks.

HTTP

Hyper Text Transfer Protocol; the World Wide Web protocol that performs the request and retrieve functions of a server

IP

Internet Protocol. A packet-based protocol for delivering data across networks**.**

IP-PBX

IP-based Private Branch Exchange

IP Telephony

(Internet Protocol telephony, also known as Voice over IP Telephony) A general term for the technologies that use the Internet Protocol's packet-switched connections to exchange voice, fax, and other forms of information that have traditionally been carried over the dedicated circuit-switched connections of the public switched telephone network (PSTN). The basic steps involved in originating an IP Telephony call are conversion of the analog voice signal to digital

format and compression/translation of the signal into Internet protocol (IP) packets for transmission over the Internet or other packet-switched networks; the process is reversed at the receiving end. The terms IP Telephony and Internet Telephony are often used to mean the same; however, they are not 100 per cent interchangeable, since Internet is only a subcase of packet-switched networks. For users who have free or fixed-price Internet access, IP Telephony software essentially provides free telephone calls anywhere in the world. However, the challenge of IP Telephony is maintaining the quality of service expected by subscribers. Session border controllers resolve this issue by providing quality assurance comparable to legacy telephone systems.

IVR

IVR is a software application that accepts a combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media.

MTU

A Maximum Transmission Unit (MTU) is the largest size packet or frame, specified in octets (eight-bit bytes), that can be sent in a packet- or frame-based network such as the Internet. The maximum for Ethernet is 1500 byte.

NAT

Network Address Translation

NTP

Network Time Protocol, a protocol to exchange and synchronize time over networks
The port used is UDP 123
Grandstream products using NTP to get time from Internet

OBP/SBC

Outbound Proxy or another name Session Border Controller. A device used in VoIP networks. OBP/SBCs are put into the signaling and media path between calling and called party. The OBP/SBC acts as if it was the called VoIP phone and places a second call to the called party. The effect of this behaviour is that not only the signaling traffic, but also the media traffic (voice, video etc) crosses the OBP/SBC. Without an OBP/SBC, the media traffic travels directly between the VoIP phones. Private OBP/SBCs are used along with firewalls to enable VoIP calls to and from a protected enterprise network. Public VoIP service providers use OBP/SBCs to allow the use of VoIP protocols from private networks with internet connections using NAT.

PPPoE

Point-to-Point Protocol over Ethernet, is a network protocol for encapsulating PPP frames in Ethernet frames. It is used mainly with cable modem and DSL services.

PSTN

Public Switched Telephone Network

i.e. the phone service we use for every ordinary phone call, or called POT (Plain Old Telephone), or circuit switched network.

RTCP

Real-time Transport Control Protocol, defined in RFC 3550, a sister protocol of the Real-time Transport Protocol (RTP), It partners RTP in the delivery and packaging of multimedia data, but does not transport any data itself. It is used periodically to transmit control packets to participants in a streaming multimedia session. The primary function of RTCP is to provide feedback on the quality of service being provided by RTP.

RTP

Real-time Transport Protocol defines a standardized packet format for delivering audio and video over the Internet. It was developed by the Audio-Video Transport Working Group of the IETF and first published in 1996 as RFC 1889

SDP

Session Description Protocol, is a format for describing streaming media initialization parameters. It has been published by the IETF as RFC 2327.

SIP

Session Initiation Protocol, An IP telephony signaling protocol developed by the IETF (RFC3261). SIP is a text-based protocol suitable for integrated voice-data applications. SIP is designed for voice transmission and uses fewer resources and is considerably less complex than H.323.
All Grandstream products are SIP based

STUN

Simple Traversal of UDP over NATs, is a network protocol allowing clients behind NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in RFC 3489. STUN will usually work good with non-symmetric NAT routers.

TCP

Transmission Control Protocol, is one of the core protocols of the Internet protocol suite. Using TCP, applications on networked hosts can create *connections* to one another, over which they can exchange data or packets. The protocol guarantees reliable and in-order delivery of sender to receiver data.

TFTP

Trivial File Transfer Protocol, is a very simple file transfer protocol, with the functionality of a very basic form of FTP; It uses UDP (port 69) as its transport protocol.

UDP

User Datagram Protocol (UDP) is one of the core protocols of the Internet protocol suite. Using UDP, programs on networked computers can send short messages known as *datagrams* to one

another. UDP does not provide the reliability and ordering guarantees that TCP does; datagrams may arrive out of order or go missing without notice. However, as a result, UDP is faster and more efficient for many lightweight or time-sensitive purposes.

VAD

Voice Activity Detection or Voice Activity Detector is an algorithm used in speech processing wherein, the presence or absence of human speech is detected from the audio samples.

VLAN

A virtual LAN, known as a VLAN, is a logically-independent network. Several VLANs can co-exist on a single physical switch. It is usually refer to the IEEE 802.1Q tagging protocol.

VoIP

Voice over IP

VoIP encompasses many protocols. All the protocols do some form of signalling of call capabilities and transport of voice data from one point to another. e.g: SIP, H.323, etc.